REMARKS

The Examiner objected to claim 4 for a typographical error, which error Applicant has corrected by changing "an" to –and-- at line 6. The Examiner also rejected claims 1 and 2 under 35 U.S.C. 103(a) as being unpatentable over Adrian et al ("Adrian") in view of Goldstein et al ("Goldstein") and objected to claims 3-7 as depending from a rejected base claim.

In contradistinction to Applicant's claimed invention Adrian discloses digital signal processing (DSP) for linearization of small input signal signals to a tri-state power switch which adds a small, fixed width, bi-state compensating carrier waveform to the leading or trailing edges of an oversampled input pulse to produce a compensated composite waveform that linearizes output from the power switch by effecting common mode cancellation of switch time errors. A correction mechanism corrects for harmonic distortion that is dependent on a modulation level or index and results from the compensating carrier modulation. The correction mechanism accesses a look-up table 66 based on the amplitude of the input signal to retrieve DSP coefficients to effect modulation distortion pre-compensation, while a harmonic compensator 64 determines and applies the inverse of the modulation-induced distortion derived from an input digital signal to the oversampled digital signal.

Likewise Goldstein deals with information transmission where an interfering signal (power supply frequency 50-60 Hz and harmonics thereof) lies within the same frequency spectrum, i.e., power supply tone compensation for voice band modems. A receiver has an equalizer 120, adaptive compensator 140 and a decision block 130 where outputs of the equalizer and the decision block are used to generate an error estimation signal for input to the adaptive compensator. The input to the decision block is generated from an output from the equalizer and a

compensating signal from the adaptive compensator, and the output of the decision block is used to generate an error adaptation signal that is a second input to the adaptation compensator and an adjustment signal for the equalizer. The adaptive compensator determines the presence of the interfering signal and estimates its frequency by subjecting the error estimation signal to a Hilbert Transform and using a phase lock loop (PLL) to generate reference signals indicative of the interfering frequencies. An adaptive combiner 166 uses the reference signals to generate a compensating signal to adjust the equalizer output. Two adaptive compensators are used – one for 50 Hz and the other for 60 Hz. Each adaptive compensator has a Hilbert Transformer/PLL 170, a harmonic ramps generator 175 and the adaptive combiner 166. The outputs from the combiners are summed (177) to generate the compensating signal. The harmonic ramps generators generate ramps for each harmonic desired, and the sine and cosine functions for each ramp feed separate adaptive combiners, the outputs of which are summed to generate the compensating signal.

Adrian adds a pre-correction to an input digital signal, while Goldstein compensates a received analog signal that already is distorted. Therefore these two references are mutually exclusive, i.e., they operate at different points in the signal path. Combining these references means that Adrian pre-correction would be applied and then Goldstein compensation applied to the pre-corrected signal after transmission and distortion. It is not clear to Applicant why one of ordinary skill in the art would use a technique for identifying and compensating for interfering signals within a frequency spectrum to pre-compensate an input signal prior to being distorted by such an interfering signal. Using Goldstein on the input signal would result in a zero compensation signal because there is no distortion (interfering signal) in the input signal. Thus there is no reason why one of ordinary skill in the art

would use the Goldstein technique in Adrian, as it would serve no purpose and not provide a pre-compensation signal.

Applicant recites in claim 1 a linearity compensation system for a non-linear system that introduces harmonic distortion into an input signal which generates from the input signal a plurality of separate corrected harmonic components using Hilbert transform filters. As the Examiner admits, Adrian does not generate a plurality of separate correction components. Further Goldstein generates harmonic components from an error estimation signal and not from the input signal. Therefore this element is not taught or suggested by Adrian and/or Goldstein.

Applicant further recites summing the corrected harmonic components with a delayed version of the input signal. Neither reference teaches or suggests combining correction components with a delayed version of the input signal, so this element also is not taught or suggested by Adrian and/or Goldstein. Thus claim 1 and all claims dependent therefrom are deemed to be allowable as being nonobvious to one of ordinary skill in the art over Adrian in view of Goldstein.

In view of the foregoing amendment and remarks allowance of claims 1-7 is urged, and such action and the issuance of this case are requested.

Respectfully submitted,

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